UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/523,923	02/07/2005	Won Yong Choi	741196-34	1911

25570 7590 12/13/2007
ROBERTS, MLOTKOWSKI & HOBBES
P. O. BOX 10064
MCLEAN, VA 22102-8064

EXAMINER	
LENNOX, NATALIE	
ART UNIT	**PAPER NUMBER**
2626	

NOTIFICATION DATE	DELIVERY MODE
12/13/2007	ELECTRONIC

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

Dbeltran@rmhlaw.com
LGallaugher@rmhlaw.com

Office Action Summary	Application No.	Applicant(s)
	10/523,923	CHOI, WON YONG
	Examiner	Art Unit
	Natalie Lennox	2626

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *February 7, 2005*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-26* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-26* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *Feb. 7, 2005* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 112

1. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

2. Claims 1, 2, 12, 14, 16, 18, and 20 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention. As per claims 1 and 12, applicant fails to describe the term "Nov," first seen in claim 1, line 6. Claims 2, 14, 16, 18, and 20, also refer to the term "Nov." For examination purposes, examiner interprets the term "Nov" as being a specific amount (number) of audio samples.

Claim Rejections - 35 USC § 102

3. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

4. Claims 1-3, 9, 11-13, and 20 are rejected under 35 U.S.C. 102(b) as being anticipated by Satyamurti (US Patent 5,806,023).

As per claim 1, Satyamurti teaches a method for time-scale modification of an audio signal by which an input signal comprised of an input stream of audio samples is

converted into an output signal modified at a desired time-scale, comprising the steps of:

determining an analysis window consisting of a first predetermined number of audio samples in said input stream (Col. 5, lines 33-35, and Col. 6, line 66 to Col. 7, line 10, also Ss of Fig. 4.);

repeating a computation of a similarity between Nov first audio samples of said analysis window and Nov second audio samples of said output signal whenever said analysis window is shifted within a predetermined search range, said similarity being calculated using third and fourth audio sample blocks consisting of audio samples down-selected from said first and second audio samples at a predetermined rate, respectively (Col. 6, lines 37-43, lines 54-65, and Col. 5, lines 57-62. The contiguous So samples are the third and fourth samples blocks); and

obtaining a shift value Km of said analysis window when a maximum value of the calculated similarity is provided (Bm from Col. 6, lines 54-65 and Col. 7, lines 11-21).

As per claim 2, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 1, further comprising the step of determining N+Nm-Nov audio samples as an add frame based upon the shift value Km and an optimal overlap length Nm at the time that a coefficient of correlation between said analysis window and said output signal is above a predetermined threshold value or provides a maximum value, said N being a value that a similarity search range Kmax between said analysis window and said output signal is deducted from said first predetermined

number (Col. 7, lines 11-21, also Col. 6, line 66 to Col. 7, line 10. The overlap length Nm is represented by the overlap segment size So.).

As per claim 3, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 2, further comprising the steps of:

forming an overlap-add block by weighting Nm audio samples from the beginning of said add frame and Nm audio samples from the end of said output signal with a weighting function (Col. 6, line 66 to Col. 7, line 10. The overlap length Nm is represented by the overlap segment size So. Also Col. 7, lines 11-21);

and substituting said overlap-add block for said Nm audio samples from the end of said output signal and adding the rest audio samples of said add frame to the end of said overlap-add block as they are (Col. 7, lines 17-21).

As per claims 9 and 13, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claims 1 and 12, further comprising the step of receiving a value α designated by a user through an input means as said desired time-scale, wherein a length ratio of said output signal to said input signal identical to said value α (Col. 5, lines 53-55, also input 218 from Fig. 2 and Col. 3, lines 47-52).

As per claim 11, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 1, wherein said similarity is determined by computing a cross- correlation (Col. 6, lines 54-58).

As per claim 12, Satyamurti teaches a method for time-scale modification of an audio signal by which an input signal comprised of an input stream of audio samples is converted into an output signal modified at a desired time-scale, comprising the steps of:

determining an analysis window consisting of N+Kmax audio samples in said input stream, where said N and said Kmax are constants (Col. 5, lines 33-35, and Col. 6, line 66 to Col. 7, line 10, also Ss of Fig. 4.);

while shifting said analysis window within a predetermined search range, computing a maximum value of a similarity between Nov audio samples of said analysis window and Nov audio samples from the end of said output signal and values of coefficient of correlation therebetween with changing said value Nov into various values (Col. 6, lines 37-43, lines 54-65, and Col. 5, lines 57-62. The contiguous So samples are the third and fourth samples blocks.);

determining N+Nm-Nov audio samples from a Km+Nov-Nmth audio sample from the beginning of said analysis window as an add frame, where said Km is a shift value of said analysis window when said maximum value of said similarity is provided, said Nm being an optimal overlap length when a coefficient of correlation between said analysis window and said output signal is above a predetermined threshold value or

provides a maximum value, and said N being a value obtained when N+Kmax is deducted by a similarity search range Kmax between said analysis window and said output signal (Col. 7, lines 11-21, also Col. 6, line 66 to Col. 7, line 10. The overlap length Nm is represented by the overlap segment size So. Bm represents the shift value Km.);

forming an overlap-add block by weighting Nm audio samples of said optimal overlap length from the beginning of said add frame and Nm audio samples of said optimal overlap length from the end of said output signal with a weighting function (Col. 6, line 66 to Col. 7, line 10. The overlap length Nm is represented by the overlap segment size So. Also Col. 7, lines 11-21); and

substituting said overlap-add block for said Nm audio samples of said optimal overlap length from the end of said output signal and simply adding the rest audio samples of said add frame to the end of said overlap-add block (Col. 7, lines 17-21).

As per claim 20, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 12, wherein said similarity between Nov audio samples of said analysis window and Nov audio samples of said output signal is determined by using a cross-correlation or said coefficient of correlation (Col. 6, lines 37-43).

Claim Rejections - 35 USC § 103

5. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

6. Claims 4-8, 16-19, and 21-26 are rejected under 35 U.S.C. 103(a) as being unpatentable over Satyamurti (US Patent 5,806,023)

As per claim 4, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 1, wherein said audio samples consisting of said third and fourth audio sample blocks have a difference in sample index as much as M1 which is a natural number bigger than 2 (Col. 5, lines 33-40).

It is noted that Satyamurti does not specifically mention that the difference in sample index is a natural number bigger than 2. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 5, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 1,

wherein said first predetermined number is N+Kmax, where N and Kmax are constants (sample size Ss from Col. 5, lines 33-40),

said search range is a range of Kmax audio samples (fixed search range of samples of the initial output block, Col. 6, lines 54-58 and lines 24-26), and

said analysis window is regularly shifted by M2 audio samples per one time shift, where M2 is a natural number bigger than 2 (Col. 6, lines 54-56).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift, where M2 is a natural number bigger than 2. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 6, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 1, wherein said audio samples consisting of said third and fourth audio sample blocks have a difference in a sample index as much as M1 which is a natural number bigger than 2, said first predetermined number being N+Kmax, where N and Kmax are constants, said search range being a range of Kmax audio samples, and said analysis window being regularly shifted by M2 audio samples per one time shift, where M2 is a natural number bigger than 2 (Col. 5, lines 33-40, and Col. 6, lines 54-58 and lines 24-26.).

It is noted that Satyamurti does not specifically mention that the difference in sample index is a natural number bigger than 2 or shifting M2 samples per one time shift, where M2 is a natural number bigger than 2. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known

options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 7, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 4, wherein said M1 being a sample index interval that is, selection interval of the audio samples consisting of said third and fourth audio sample blocks has a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size (Col. 5, lines 33-40).

As per claim 8, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 4, further comprising the step of preparing corresponding values each of which is mapped into each one of various sampling rates of audio signals in advance and applying a corresponding value mapped at a sampling rate figured out from header information of said input signal as an assigned value of said M1 being a sample index interval that is, selection interval of the audio samples consisting of said third and fourth audio sample blocks (Ss from Col. 5, lines 33-40 and So (3^{rd} and 4^{th} audio sample blocks) from Col. 5, lines 41-47 and lines 57-62).

As per claim 16, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 12, wherein audio samples participated in computing said similarity and said coefficient of correlation are selected among signals belonging to the respective Nov audio samples of said analysis window and said output signal and adjacent audio samples of said participated audio samples have a difference in sample index as much as M1 which is a natural number bigger than 2 (So (Nov samples) Col. 6, lines 37-44 and Col. 5, lines 33-40).

It is noted that Satyamurti does not specifically mention that the difference in sample index is a natural number bigger than 2. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 17, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 12, wherein said shifting of said analysis window is performed in a manner that said analysis window is regularly shifted by M2 audio samples per one time shift, where M2 is a natural number bigger than 2 and the number of shifted audio samples in total is not larger than Kmax audio samples of a search range (Col. 6, lines 54-58 and lines 24-26).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift, where M2 is a natural number bigger than 2. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 18, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 12, wherein audio samples participated in computing said similarity and said coefficient of correlation are selected among signals belonging to the respective Nov audio samples of said analysis window and said output signal, adjacent audio samples of said participated audio samples having a difference in sample index as much as M1 which is a natural number bigger than 2, said shifting of said analysis window being performed in a manner that said analysis window is regularly shifted by M2 audio samples per one time shift, where M2 is a natural number bigger than 2, and the number of shifted audio samples in total being not larger than Kmax audio samples of a search range (So (Nov samples) Col. 6, lines 37-44 and Col. 5, lines 33-40, also . 6, lines 54-58 and lines 24-26).

It is noted that Satyamurti does not specifically mention that the difference in sample index is a natural number bigger than 2 or shifting M2 samples per one time shift, where M2 is a natural number bigger than 2. However, the claim would have been

obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 19, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 16, wherein said parameter M1 has a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size (Col. 5, lines 33-40).

As per claim 21, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 5, wherein said M2 being a shift interval of said analysis window has a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size (Col. 5, lines 33-40, also Col. 6, lines 54-56).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 22, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 6, wherein said M1 being a sample index interval, that is, selection interval, of the audio samples consisting of said third and fourth audio sample blocks and said M2 being a shift interval of said analysis window have a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size, respectively (Col. 5, lines 33-40, also Col. 6, lines 54-56).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 23, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 5, further comprising the step of preparing corresponding values each of which is mapped into each one of various sampling rates of audio signals in advance and applying a corresponding value mapped at a sampling rate figured out from header information of said input signal as an assigned value of said M2 being a shift interval of said analysis window (Col. 5, lines 33-40, also Col. 6, lines 54-56).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 24, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 6, further comprising the step of preparing corresponding values each of which is mapped into each one of various sampling rates of audio signals in advance and applying a corresponding value mapped at a sampling rate figured out from header information of said input signal as an assigned value of said M1 being a sample index interval, that is, selection interval, of the audio samples consisting of said third and fourth audio sample blocks and/or said M2 being a shift interval of said analysis window (Col. 5, lines 33-40, also Col. 6, lines 54-56).

It is noted that Satyamurti does not specifically mention shifting M2 samples per one time shift. However, the claim would have been obvious because "a person of ordinary skill has good reason to pursue the known options within his or her technical grasp. If this leads to the anticipated success, it is likely the product not of innovation but of ordinary skill and common sense." An anticipated success may be a reduction in computation time by processing fewer samples.

As per claim 25, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 17, wherein said parameter M2 has a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size (Col. 5, lines 33-40).

As per claim 26, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 18, wherein said parameter M1 and said parameter M2 have a value of one of two integers closest to a value obtained by dividing an actual sampling rate of said input signal by a reference sampling rate of a predetermined size, respectively (Col. 5, lines 33-40).

7. Claims 10 and 14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Satyamurti (US Patent 5,806,023) in view of Hejna, Jr. et al. (US Patent 5,175,769).

As per claims 10 and 14, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claims 7 and 12, but does not specifically mention wherein a first audio sample of a mth analysis window is an mSath audio sample from the beginning of said input stream, and said value Nov being reduced at a predetermined rate by setting N-Ss as a maximum value thereof, where said Ss is a fixed value, and said Sa is determined by a relation of Ss= α Sa.

However, Hejna, Jr. et al. teaches a first audio sample of a mth analysis window is an mSath audio sample from the beginning of said input stream, and said value Nov being reduced at a predetermined rate by setting N-Ss as a maximum value thereof, where said Ss is a fixed value, and said Sa is determined by a relation of Ss= α Sa (Col. 15, lines 35-40, and Col. 5, lines 10-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have used the feature of a first audio sample of a mth analysis window is an mSath audio sample from the beginning of said input stream, and said value Nov being reduced at a predetermined rate by setting N-Ss as a maximum value thereof, where said Ss is a fixed value, and said Sa is determined by a relation of Ss= α Sa as taught by Hejna, Jr. et al. for Satyamurti's method because by having a fixed region of overlap the number of computations which are required to evaluate the similarity measure over the range of shift values are reduced over that required with an unfixed region of overlap (Hejna, Jr.'s Col. 5, lines 20-25).

8. Claim 15 is rejected under 35 U.S.C. 103(a) as being unpatentable over Satyamurti (US Patent 5,806,023) in view of Sabin (Computer Map Typing - Optimizing the Correlation Coefficient Threshold, Jan 1974).

As per claim 15, Satyamurti teaches a method for time-scale modification of an audio signal as claimed in claim 12, but does not specifically mention wherein said threshold value with respect to said coefficient of correlation is over 0.7.

However, Sabin teaches said threshold value with respect to said coefficient of correlation is over 0.7 (Abstract).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have used the feature of said coefficient of correlation is over 0.7 as taught by Sabin for Satyamurti's method because Sabin provides a correlation coefficient threshold value of 0.8 in order to obtain a reasonable degree of similarity (Abstract).

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Natalie Lennox whose telephone number is (571) 270-1649. The examiner can normally be reached on Monday to Friday 9:30 am - 7 pm (EST).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

NL 12/06/2007

RICHEMOND DORVIL
SUPERVISORY PATENT EXAMINER